Akuvox

R23X Door Phone User Manual

About This Manual

Thank you for choosing Akuvox's products. In user manual, we provides all functions and configurations you want to know about R23X. Please verify the packaging content and network status before setting. This manual applies to firmware 26.0.0.96 or lower version.

Note: The old firmware may be a little different from 26.0.0.96 about some configuration. Please consult your administrator for more information.

Content

1.Overview

1.1. Product Description

R23C

R23P

Akuvox's Audio Doorphone R23X is an open, non-proprietary and IP-based door station for two-way communication and remote entry control. It is a perfect complement to any SIP system and offers new possibilities of effectively control entry to your premises.It’s applicable in villas, office and so on.

FCC Caution:

Any Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

Note: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

—Reorient or relocate the receiving antenna.

—Increase the separation between the equipment and receiver.

—Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.

—Consult the dealer or an experienced radio/TV technician for help.

Specific Absorption Rate (SAR) information
SAR tests are conducted using standard operating positions accepted by the FCC with the device transmitting at its highest certified power level in all tested frequency bands, although the SAR is determined at the highest certified power level, the actual SAR level of the device while operating can be well below the maximum value. Before a new product is a available for sale to the public, it must be tested and certified to the FCC that it does not exceed the exposure limit established by the FCC, tests for each phone are performed in positions and locations as required by the FCC. For headset, this part has been tested and meets the FCC RF exposure guidelines when used with an accessory designated for this product or when used with an accessory that contains no metal.
For baseband, this equipment complies with FCC radiation exposure limits set forth for an uncontrolled environment .This equipment should be installed and operated with minimum distance 20cm between the radiator& your body.

1.2. Feature

➢ Highlight

- Vandal resistant body, with a flush button
- POE(IEEE802.3af, Power-over-Ethernet)
- Two-way audio communication over IP network with Echo cancel feature
- Complied with SIP Standard for easy integration in each SIP PBXes

➢ Physical&Power

- Body material: all-aluminum
- Button: 1 call button
- Output Relay: 2 output relays for door opener
- 802.3af Power-Over-Ethernet
- 12 DC connector(if not using POE)
- RF Card Reader:13.56MHz Supported (R23C only)
- Power consumption: less than 12w
- Water proof&Dust proof: IP65
- Installation: Wall-mounted
- Dimension: 190x110x35mm

➢ SIP Endpoint

- SIP v1(RFC2543), SIP v2(RFC3261)
- Audio codecs: G.711a, G.711μ, G.722, G.729
- Speech Quality: 7kHz Audio
- Echo Cancellation
- Voice Activation Detection
- Comfort Noise Generator
- Door opened via DTMF post-dial

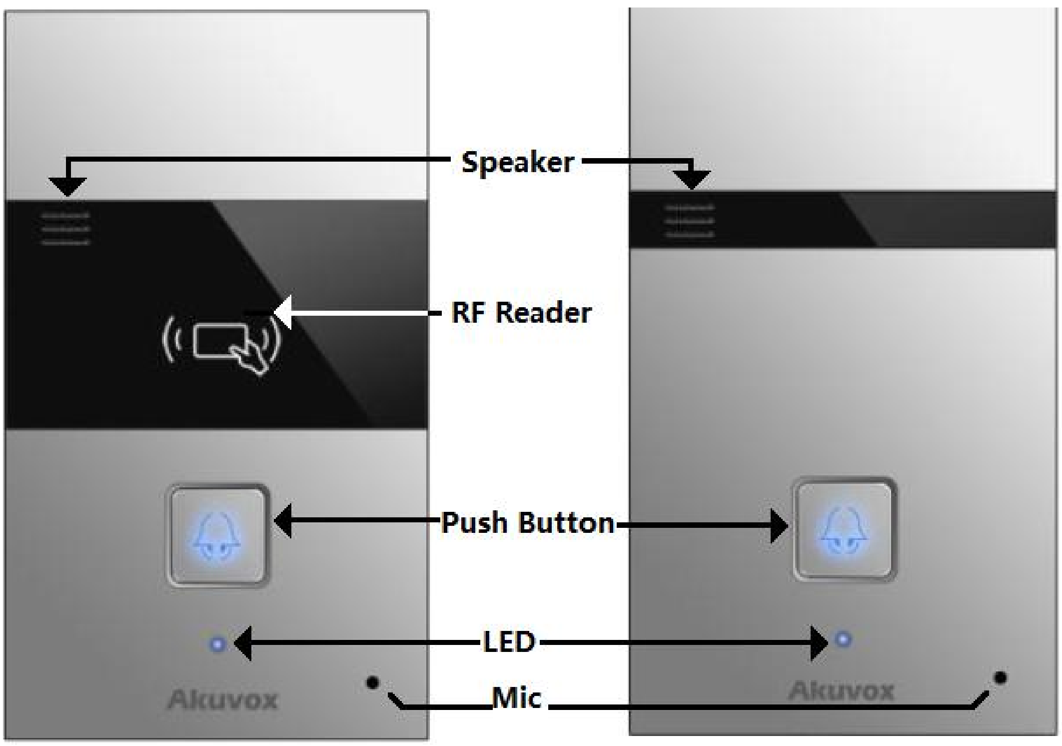
➢ **Network Features**

- 1x10/100Mbps Ethernet Port
- Protocols support: IPv4, HTTP, HTTPS, FTP, SNMP, DNS, NTP, RTSP, RTP, TCP, UDP, ICMP, DHCP, ARP

➢ **Application Scenarios**

- Office door phone with on-site or hosted IP-PBX
- Remote site entry over Internet
- Villa intercom with door access control

1.3. Panel Description



R23C **R23P**

2. Configuration

2.1. Web login

2.1.1 Obtain the IP address

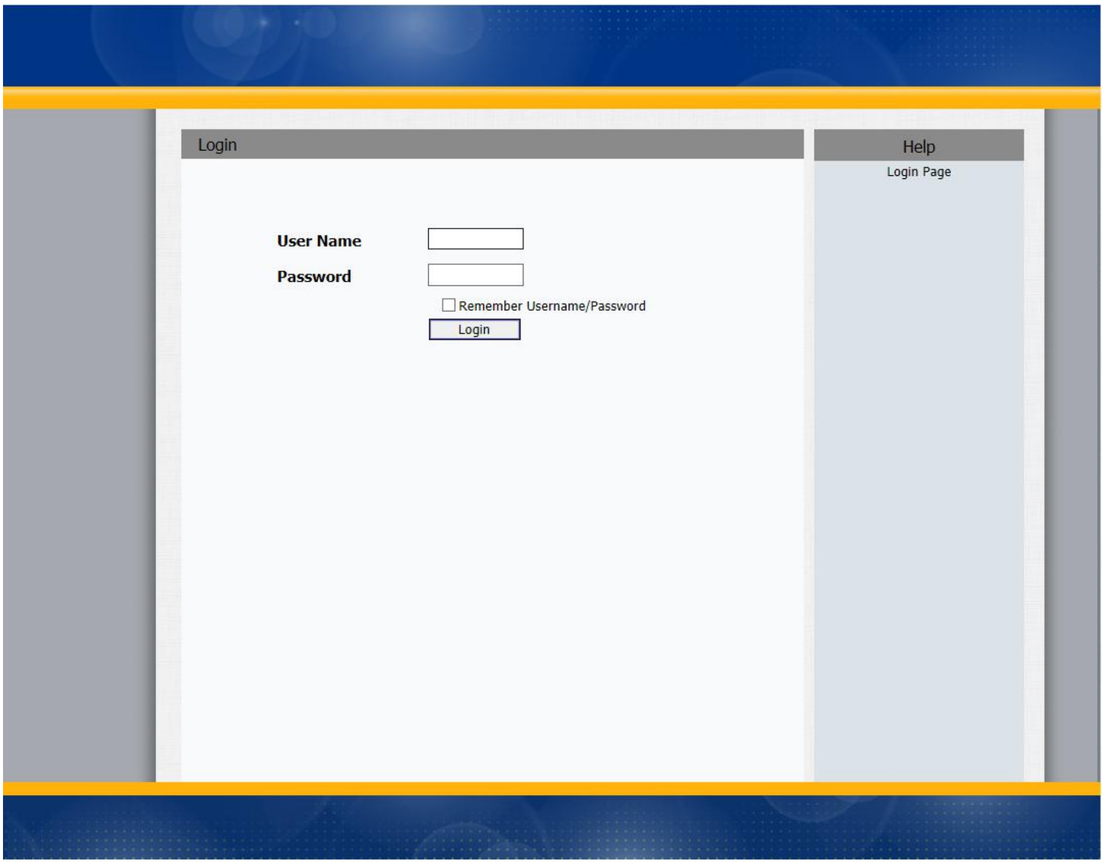
The Akuvox R23X uses Static IP by default, the default IP address is 192.168.1.100. If the IP address is unknown, press and hold the call button for a short period of time(about 5s) after LED light turns blue, the phone will announce its IP.

2.1.2 Login the web

Open a Web Browser, enter the corresponding IP address. Then, type the default user name and password as below to log in:

User name: admin

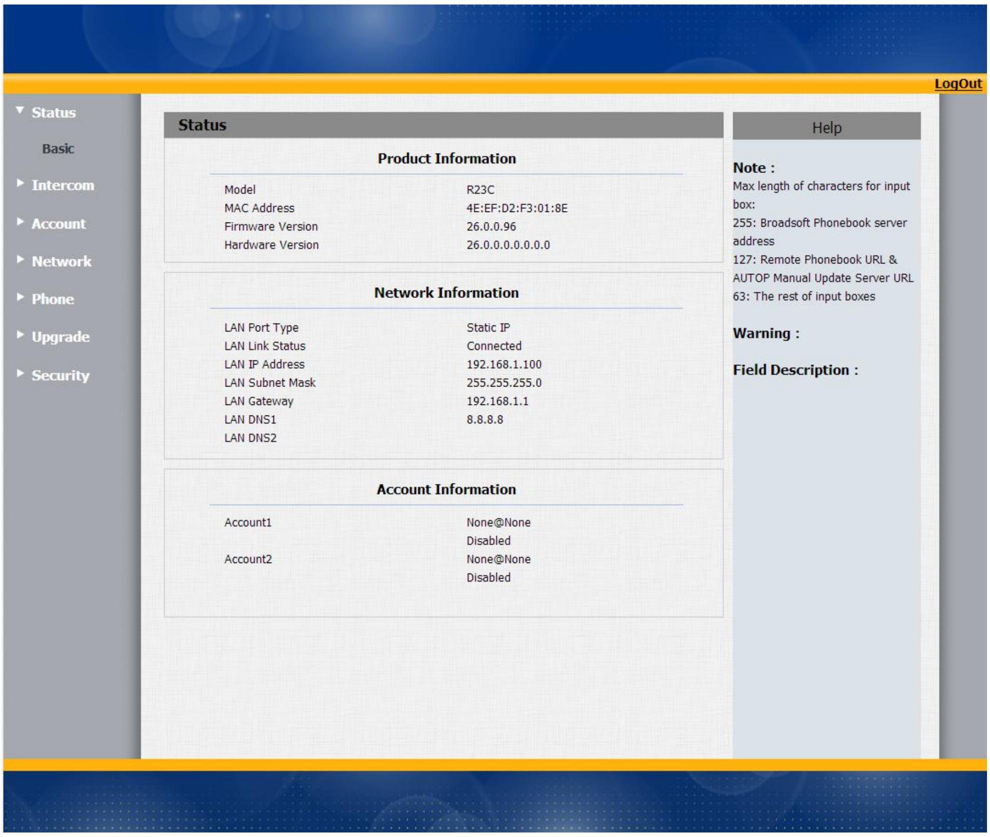
Password: admin



2.2. Status

2.2.1 Basic

Status can be viewed from “Status -> Basic”,including the information of product, network and account.

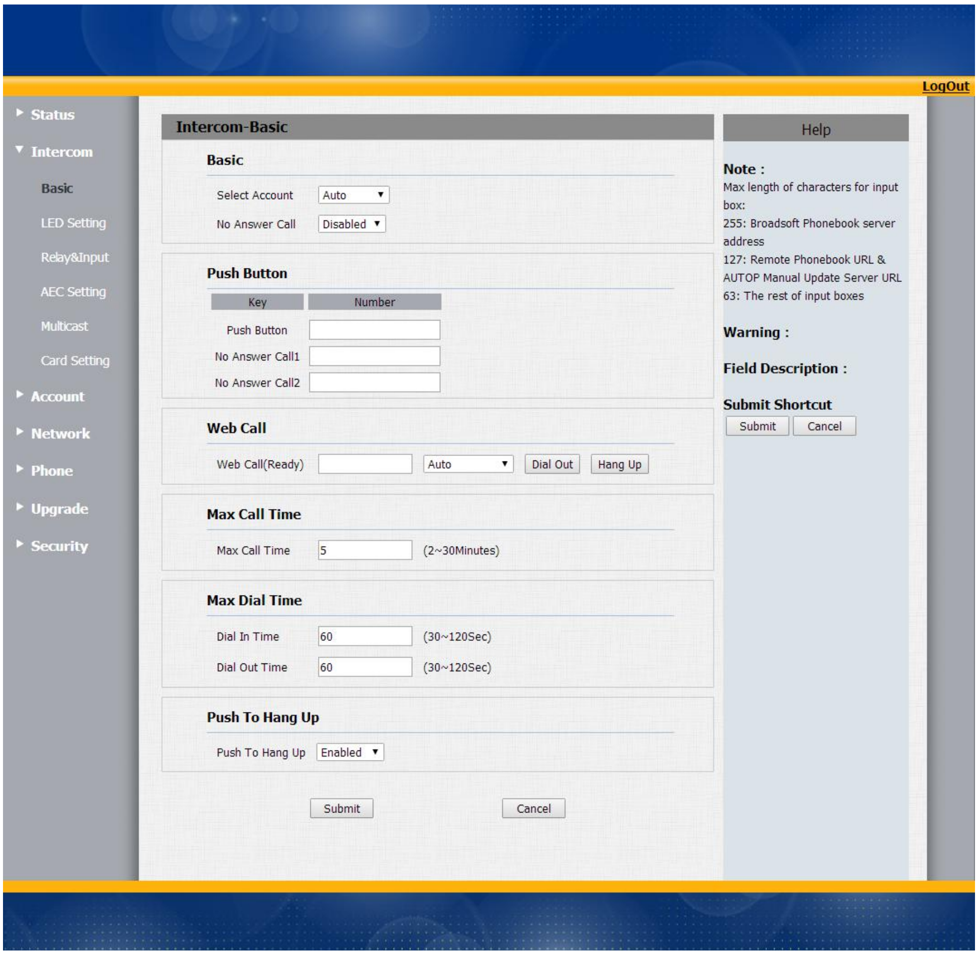


Sections	Description
Product Information	To display the device’s information such as Model name, MAC address (IP device’s physical address), Firmware version and Hardware version.
Network Information	To display the device’s Networking status(LAN Port),such as Port Type(which could be DHCP/Static/PPPoE), Link Status, IP Address, Subnet Mask, Gateway, Primary DNS server, Secondary DNS server, Primary NTP server and Secondary NTP server(NTP server is used to synchronize time from INTERNET automatically).
Account Information	To display device’s Account information and Registration status (account’s username, registered server’s address, Register result).

2.3. Intercom

2.3.1 Basic

Go to the path: Intercom-Basic

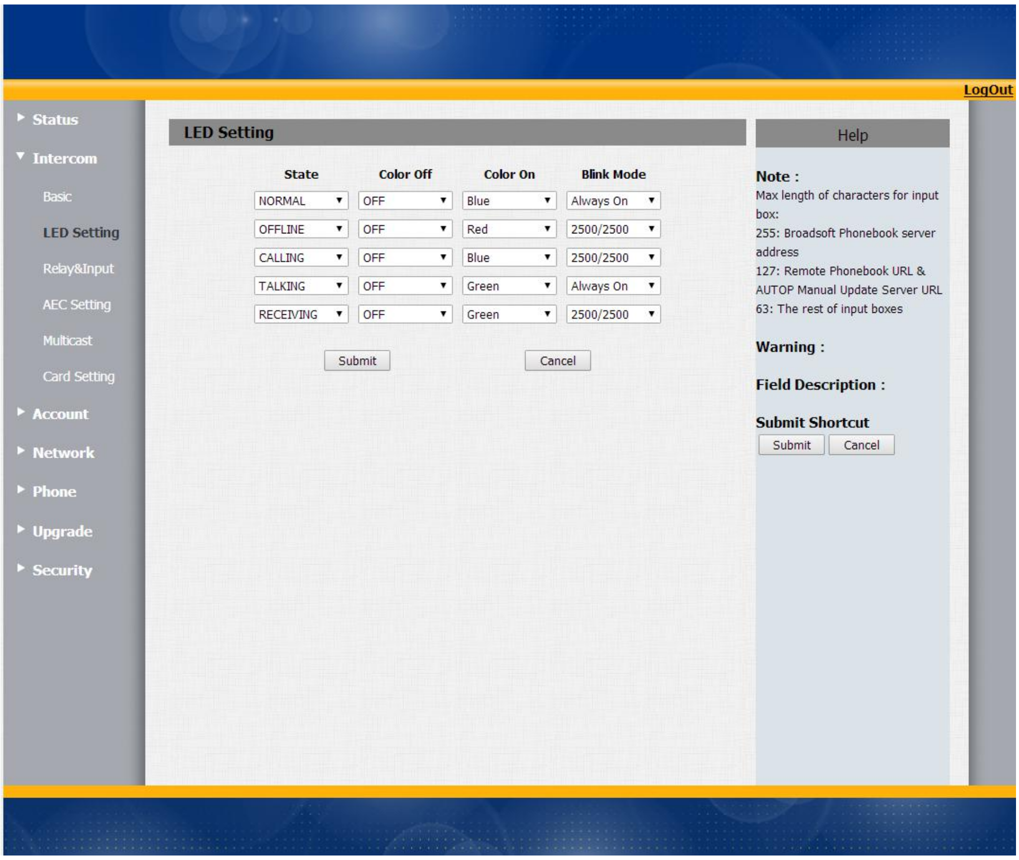


Sections	Description
Basic	● Select Account: R23X supports 2 accounts. You can choose one account or Auto mode for the following Intercom basic settings. ● No Answer Call： R23X will call to the No answer call number in order when the ringtone is time out without answer of the push button number. Disable by default.
Push Button	● Push Button: To configure the destination number or IP you want to contact with. ● No Answer Call 1&2: To setup two no answer call numbers or one no answer call number.
Web Call	To dial out or answer the phone from website.

Max Call Time	To configure the max call time
Max Dial Time	● Dial in Time: When other phone calls to R23X, if ring tone is over the Dial in Time without answer. The call will be hang up. ● Dial out Time: When R23X calls to the other party, if the ringtone is over the Dial out Time without answer. R23X will continue calls to no answer call number in order.
Push to Hang up	To enable or disable the Push to Hang up function

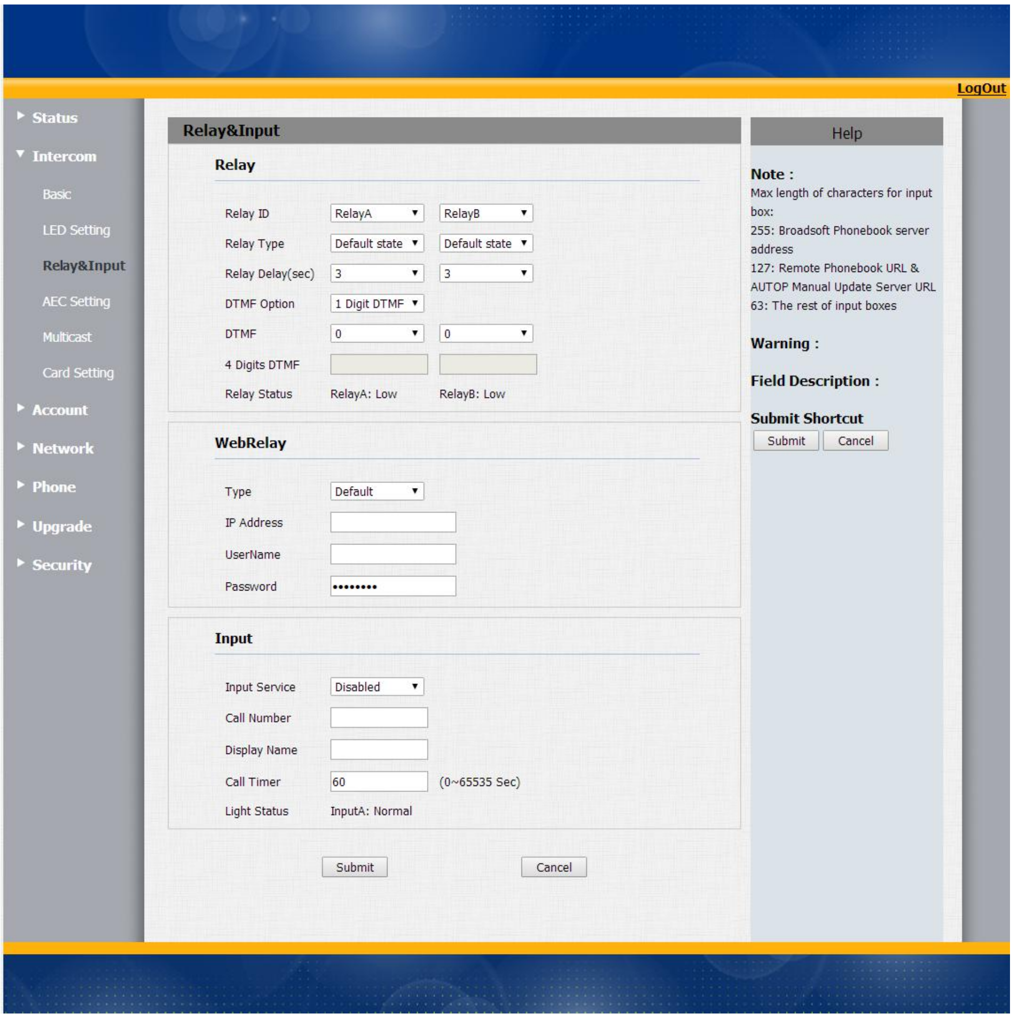
2.3.2 LED Setting

To setup the LED lighting mode.



Sections	Description
State	Including five states:Normal,Offline,Calling,Talking and Receiving.
Color Off	The default status is OFF.
Color On	It can support three color: Red, Green, Blue.
Blink Mode	To setup the different blink frequency.

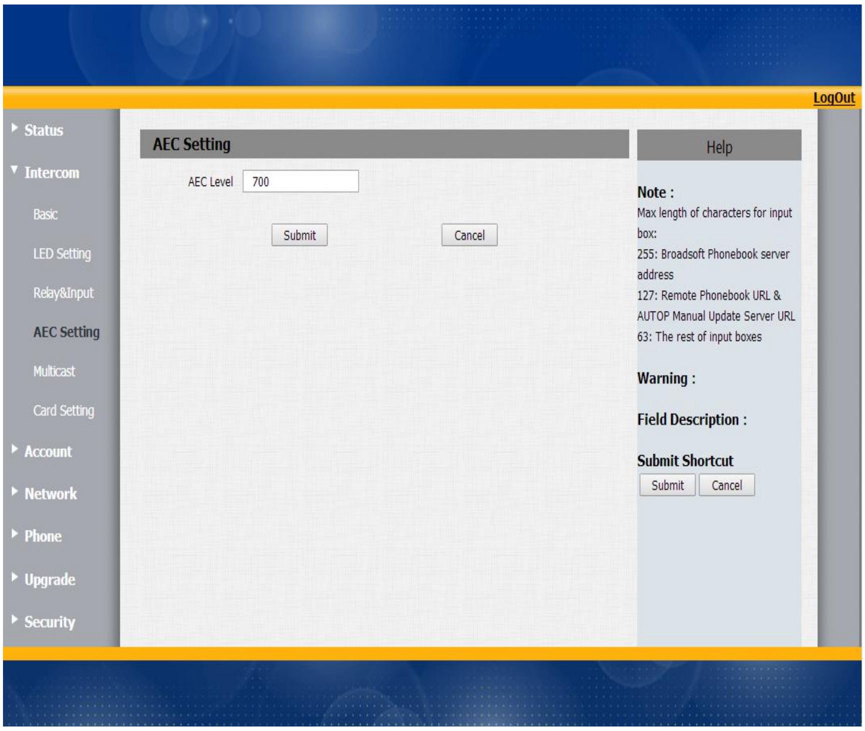
2.3.3 Relay&Input



Sections	Description
Relay	To configure some settings about unlock ● Relay Select: R23X supports 2 relays. ● Relay Type: Different locks use different relay types, positive or negative. If you connect the Lock in NO connector, select positive type. Otherwise using negative type. ● Relay Delay(sec): Allows the door to remain “open” for certain period . The range is from 1 to 5 seconds. ● DTMF Option: R23X support 1digit or 4 digits DTMF unlock code. Please select one type and enter the corresponding code. ● DTMF: Setup 1 digit DTMF code for remote unlock ● 4 Digits DTMF : Setup 4 digits DTMF code for remote unlock. ● Status: Different relay types will show different status.

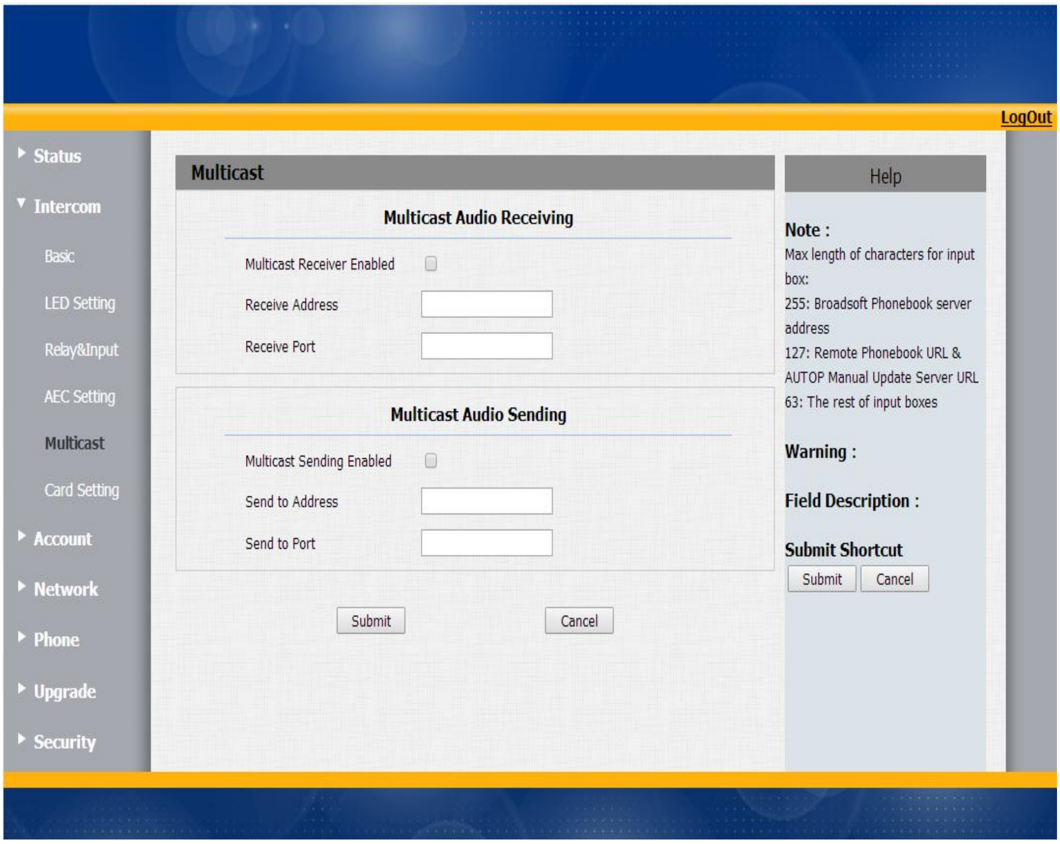
Web relay	R23X can support extra web relay. This function is more safety to use DTMF code to remote unlock. ● Type: Connect web relay and choose the type. ● IP Address: Enter web relay IP address. ● User name: it is an authentication for connecting web relay ● password: it is an authentication for connecting web relay **Note:** Users can modify username and password in web relay website.
Input	There is a sensor that used for anti-vandal in R23X. When R23X is broken by violent means, the sensor will be triggered, then the management center will receive the alarm. ● Service: Enable by default ● Call Number: To setup management center number for alarm. ● Display Name: Which is sent to the other call party for displaying. ● Call Timer: The interval of calling. For instant , the Call timer is 5sec, if you hang up the calling in the third second, the calling will auto call out after 2sec. ● Light Status: The status will change according to the sensor. Once the sensor is triggered , the status will show Warning. Normal by default.

2.3.4 AEC Setting



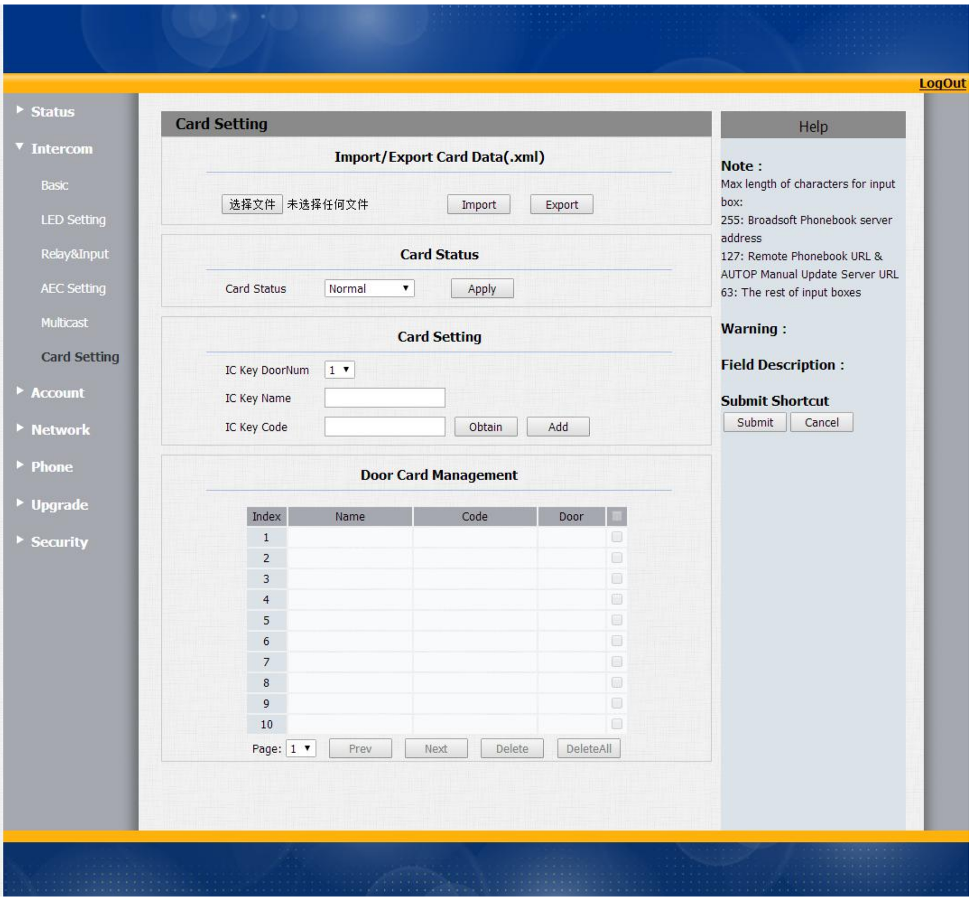
Sections	Description
AEC Level	AEC(Configurable Acoustic and Line Echo Cancelers) is used to adjust the echo effect during the communication. The default value is 700. Increase the level, the echo control is better.

2.3.5 Multicast



Sections	Description
Multicast Audio Receiving	To display and configure the Multicast setting. ● Multicast Receiver Enable: Enable receiver multicast function. ● Receiver address : Setup the multicast address. ● Receiver port : setup the multicast address port.
Multicast Audio Sending	To setup the multicast parameters. ● Multicast Sending Enable: Enable sender multicast function ● Send to Address: setup the multicast address. ● Send to port: setup the multicast address port.

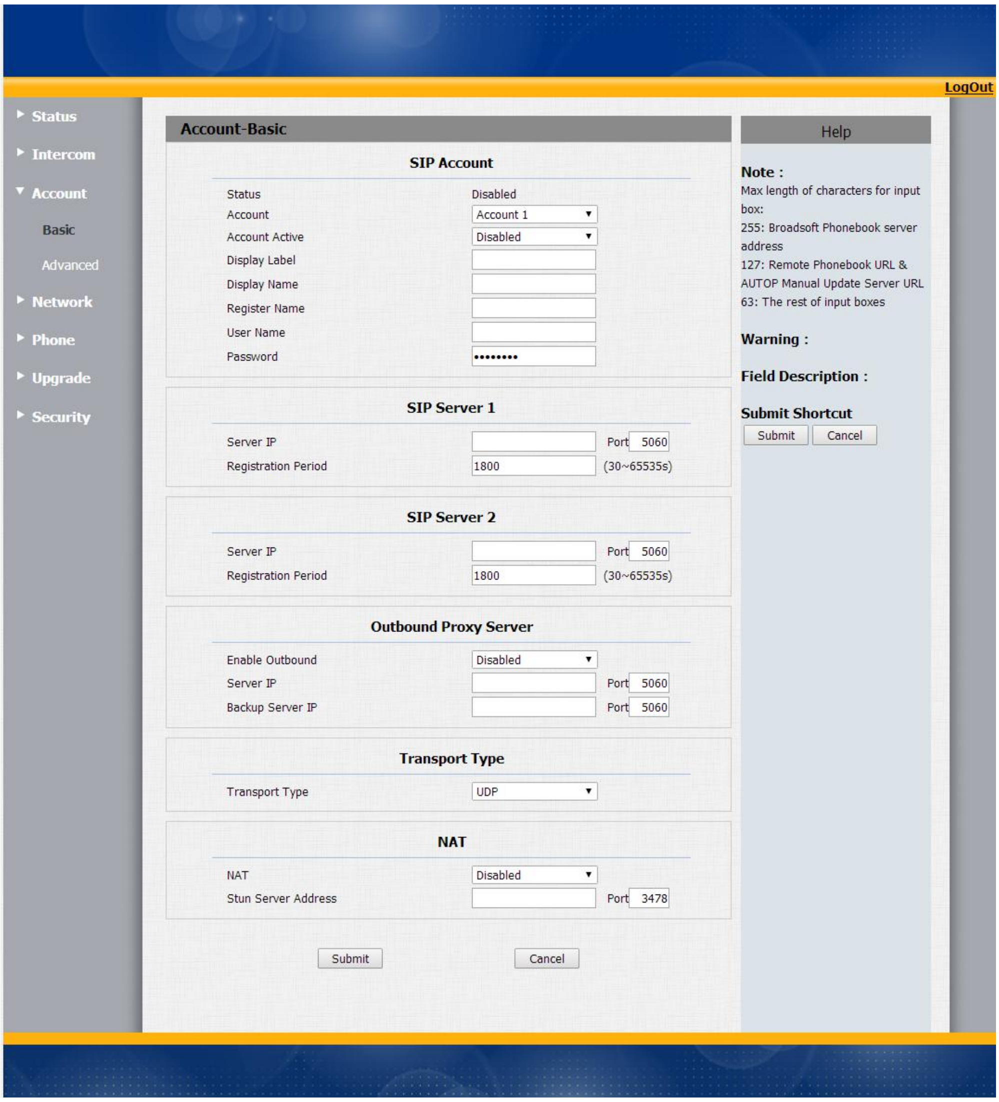
2.3.6 Card Setting(R23C only)



Sections	Description
Import/Export Card Data	To import or export the card data file. Only support .xml format.
Card Status	● Normal: choose Normal mode when reading card. ● Card Issuing: Choose Card Issuing mode when writing card
Card Setting	● IC Key DoorNum: R23X can support to connect 2 doors. Choose one and add the valid card for unlock. ● IC Key Name: To setup corresponding name for the card. ● IC Key Code: Place the card in the Card-reading area, click “obtain” to read the card code, click “Add” and the card information will show in the Door Card Management list.
Door Card Management	Valid card information will show in the list. Users can tick the current card information then delete one or all in the list.

2.4. Account

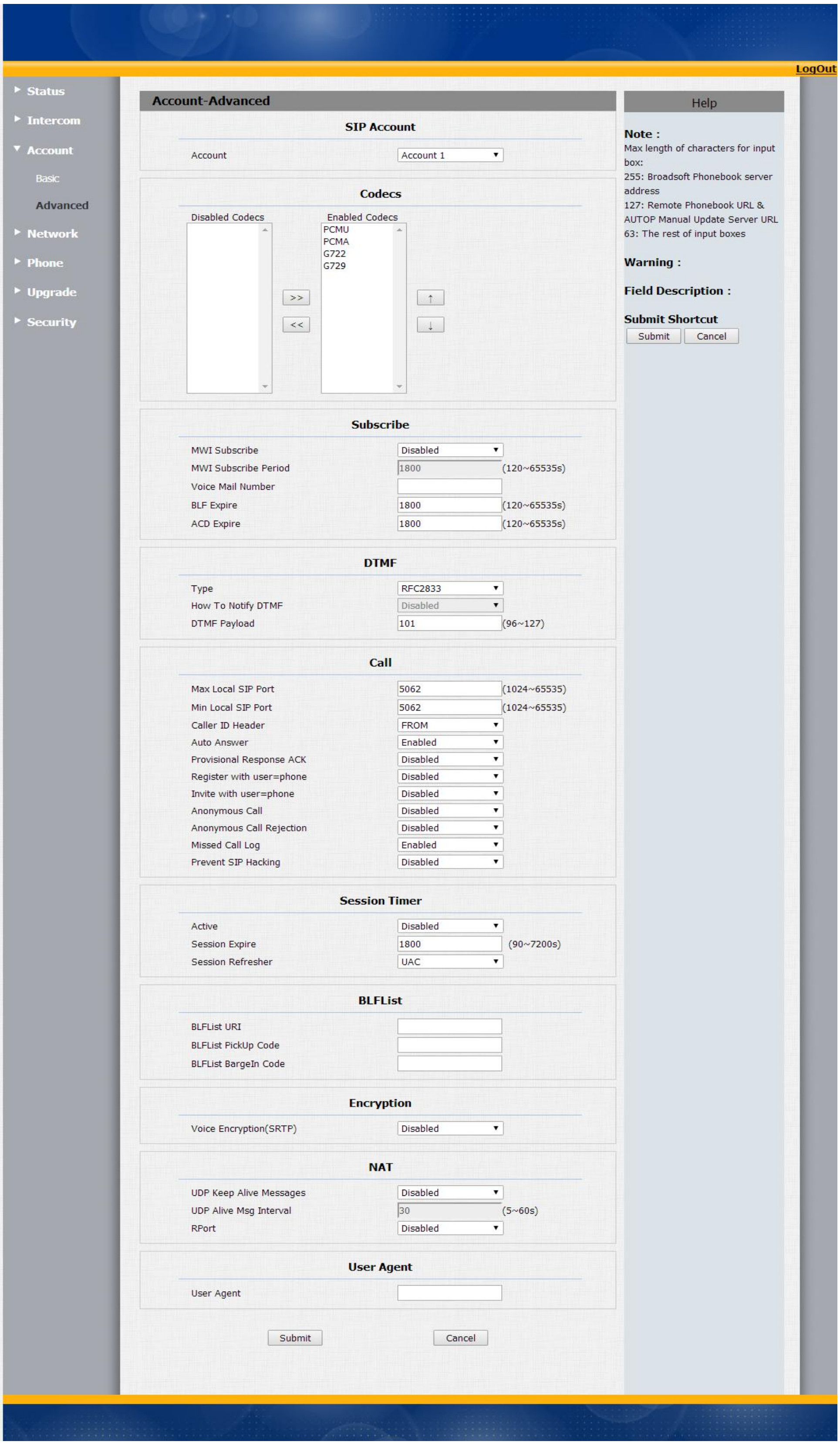
2.4.1 Basic



Sections	Description
SIP Account	To display and configure the specific Account settings. ● Status: To display register result. ● Display Name: Which is sent to the other call party for display. ● Register Name: Allocated by SIP server provider, used for authentication. ● User Name: Allocated by your SIP server provide, used for authentication. ● Password: Used for authorization.

SIP Server 1	To display and configure Primary SIP server settings. ● Server IP: SIP server address, it could be an URL or IP address. ● Registration Period: The registration will expire after Registration period, the IP phone will re-register automatically within registration period.
SIP Server 2	To display and configure Secondary SIP server settings. Used for redundancy, if registering to Primary SIP server fails, the IP phone will go to Secondary SIP server for registering. **Note**: Secondary SIP server is used for redundancy, it can be left blank if there is not redundancy SIP server in user's environment.
Outbound Proxy Server	To display and configure Outbound Proxy server settings. An outbound proxy server is used to receive all initiating request messages and route them to the designated SIP server. **Note**: All SIP request messages from the IP phone will be sent to the outbound proxy server forcefully when configured.
Transport Type	To display and configure Transport type for SIP message ● UDP: UDP is an unreliable but very efficient transport layer protocol. ● TCP: Reliable but less-efficient transport layer protocol. ● TLS: Secured and Reliable transport layer protocol. ● DNS-SRV: A DNS RR for specifying the location of services.
NAT	To display and configure NAT(Net Address Translator) settings. ● STUN: Simple Traversal of UDP over NATS is a solution to solve NAT issues. **Note**: By default, NAT is disabled.

2.4.2 Account-Advanced

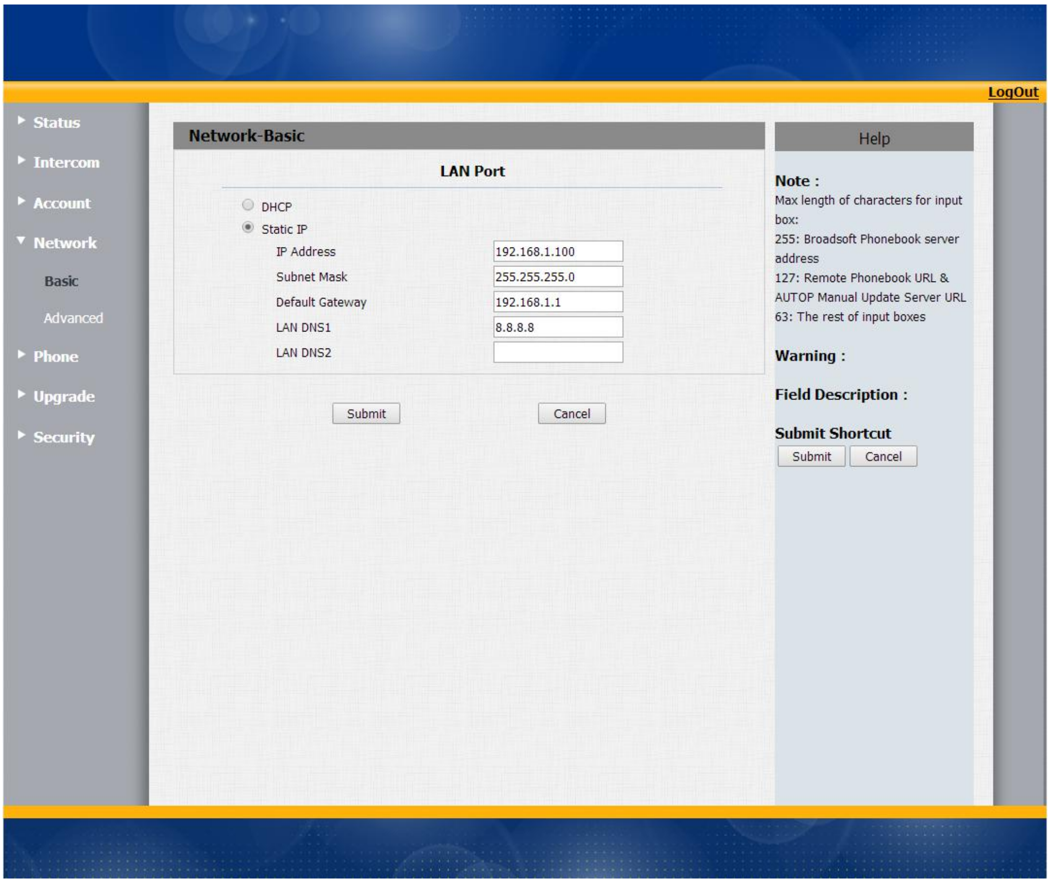


Sections	Description
SIP Account	Select an account to display the settings.
Codecs	To display and configure available/unavailable codecs list. Codec means coder-decoder which is used to transfer analog signal to digital signal or vice versa. Familiar codecs are PCMU(G711U), PCMA(G711A), G722 (wide-bandth codecs), G729 and so on.
Subscribe	To display and configure MWI, BLF, ACD subscription settings. ● MWI: Message Waiting Indicator which is used to indicate whether there is unread new voice message. ● BLF: BLF is short for Busy Lamp Field which is used to monitor the designated extension status. ● ACD: Automatic Call Distribution is often used in offices for customer service, such as call center. The setting here is to negotiate with the server about expire time of ACD subscription.
DTMF	To display and configure DTMF settings. ● Type:Support Inband,Info,RFC2833 or their combination. ● How To Notify DTMF: Only available when Type is Info. ● DTMF Payload: To configure payload type for DTMF. **Note**: Type RFC2833 is set by default as a standard. Type Inband uses inband frequency to indicate DTMF tone which is most used to be compatible to traditional telephone server. Type Info use SIP Info message to indicate DTMF message.
Call	To display and configure call-related features. ● Max Local SIP Port: To configure maximum local sip port for designated account. ● Min Local SIP Port: To configure minimum local sip port for designated account. ● Caller ID Header: To configure which Caller ID format to fetch for displaying on Phone UI. ● Auto Answer: IP phone will answered the incoming call for designated account automatically when enabled. ● Ringtones: Choose the ringtone for each account. ● Provisioning Response ACK: 100% reliability for all provisional messages, this means it will send ACK every time when the IP phone receives a provisional SIP message from SIP server. ● User=phone: If enabled, IP phone will send user=phone within SIP message. ● PTime: Interval time between two consecutive RTP packets. ● Anonymous Call: If enabled, all outgoing call to the designated account will be anonymous number.

	• Anonymous Call Rejection: If enabled, all incoming anonymous-out call of the designated account will be rejected. • Missed Call Log: To display the miss call log. • Prevent SIP Hacking: Enable to prevent SIP from hacking.
Session Timer	To display or configure session timer settings. • Active: To enable or disable this feature. If enable, the on going call will be disconnected automatically once the session expired unless it's been refreshed by UAC or UAS. • Session Expire: Configure session expire time. • Session Refresher: To configure who should be response for refreshing a session. **Note**: UAC means User Agent Client, here stands for IP phone. UAS means User Agent Server, here stands for SIP server.
BLF List	To display or configure BLF List URI address. • BLF List URI: BLF List is short for Busy Lamp Field List. • BLF List Pick Up Code: To set the BLF pick up code. • BLF List Barge In Code : To set the BLF barge in code.
Encryption	To enable or disabled SRTP feature. • Voice Encryption(SRTP): If enabled, all audio signal (technically speaking it's RTP streams) will be encrypted for more security.
NAT	To display NAT-related settings. • UDP Keep Alive message: If enabled, IP phone will send UDP keep-alive message periodically to router to keep NAT port alive. • UDP Alive Msg Interval: Keep alive message interval. • Rport: Remote Port, if enabled, it will add Remote Port into outgoing SIP message to designated account.
User Agent	One can customize User Agent field in the SIP message; If user agent is set to specific value, user could see the information from PCAP. If user agent is not set by default, user could see the company name, model number and firmware version from PCAP

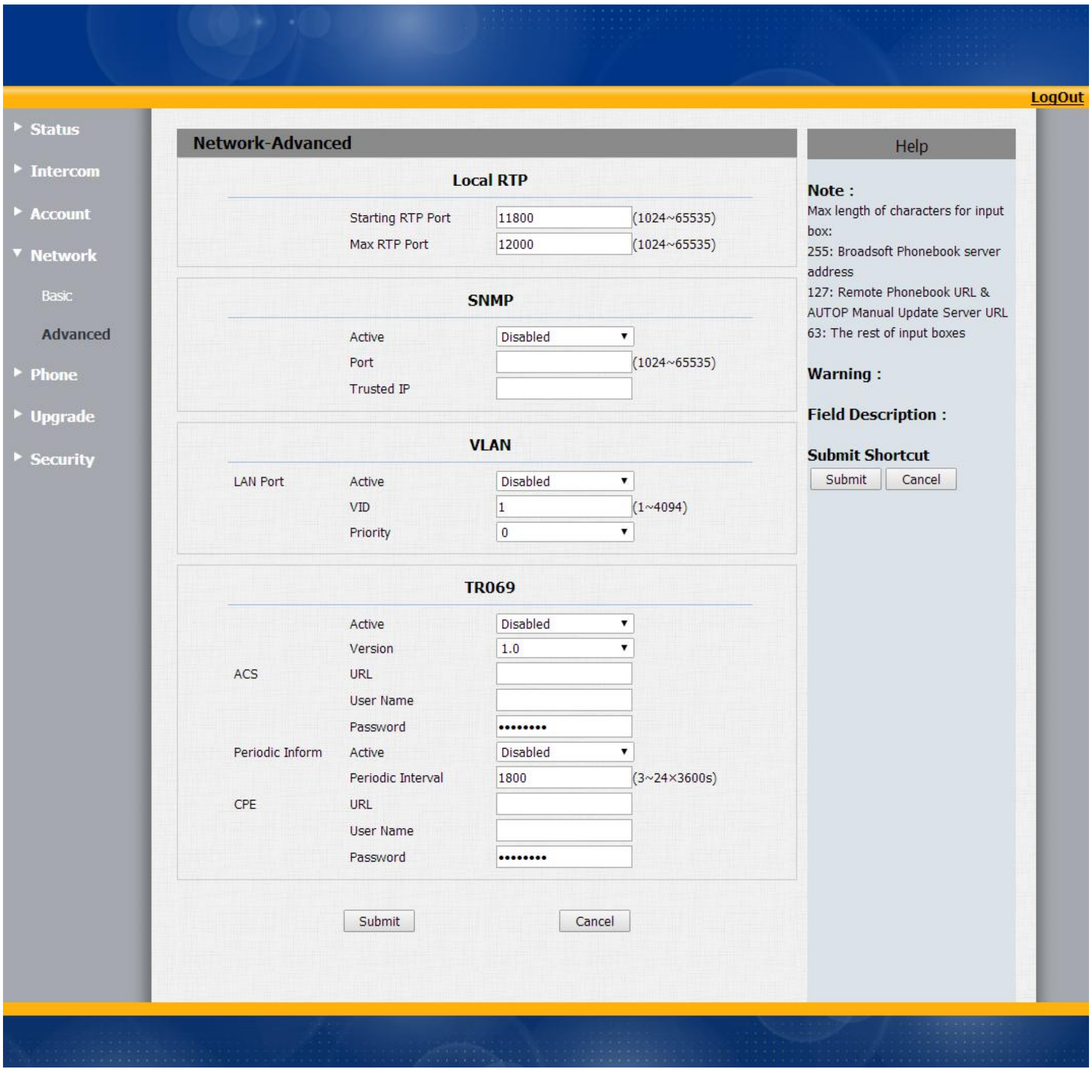
2.5. Network

2.5.1 Basic



Sections	Description
LAN Port	To display and configure LAN Port settings. ● DHCP: If selected, IP phone will get IP address, Subnet Mask, Default Gateway and DNS server address from DHCP server automatically. ● Static IP: If selected, you have to set IP address, Subnet Mask, Default Gateway and DNS server manually.

2.5.2 Advance

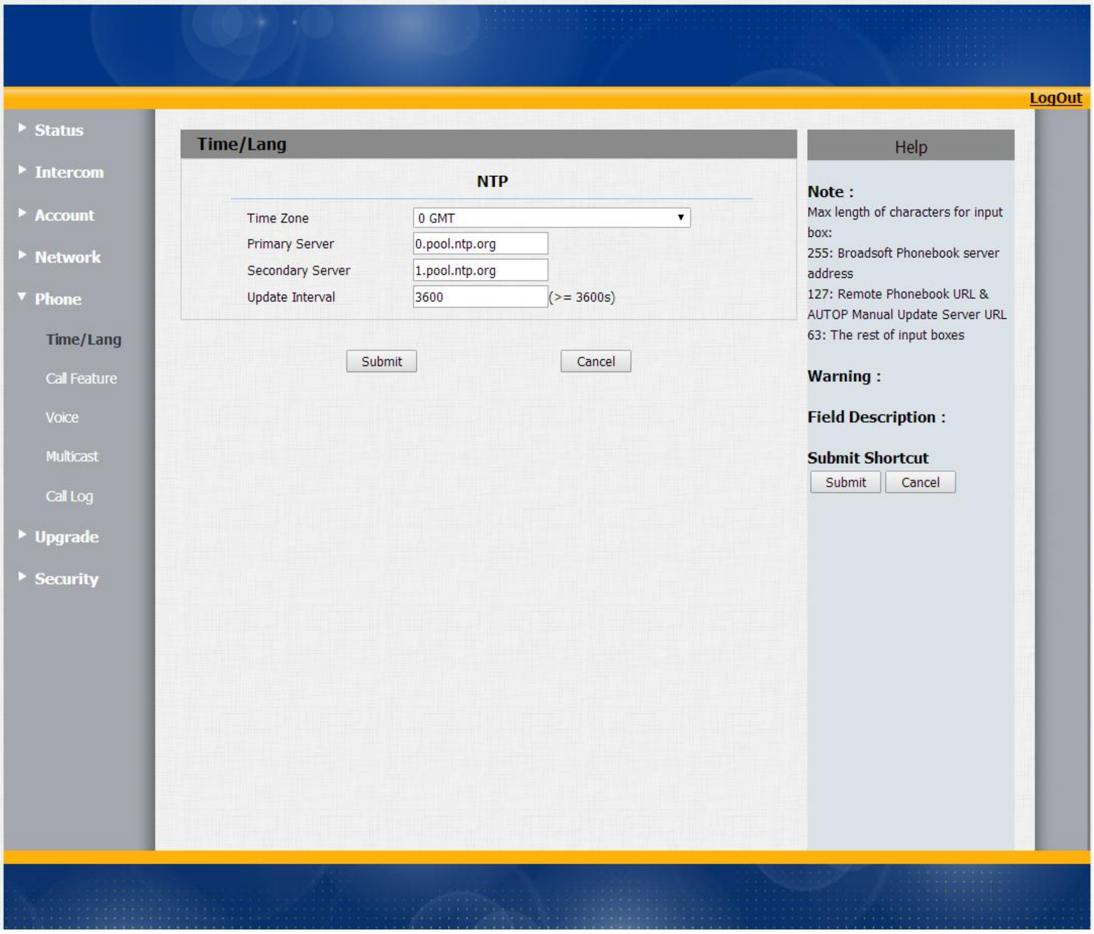


Sections	Description
Local RTP	To display and configure Local RTP settings. ● Max RTP Port: Determine the maximum port that RTP stream can use. ● Starting RTP Port: Determine the minimum port that RTP stream can use.
SNMP	To display and configure SNMP settings. ● Active: To enable or disable SNMP feature. ● Port: To configure SNMP server’s port. ● Trusted IP: To configure allowed SNMP server address, it could be an IP address or any valid URL domain name. **Note**: SNMP (Simple Network Management Protocols) is an Internet-standard protocol for managing devices on IP networks.

VLAN	To display and configure VLAN settings. ● Active: To enable or disable VLAN feature for designated port. ● VID: To configure VLAN ID for designated port. ● Priority: To select VLAN priority for designated port. **Note**: Please consult your administrator for specific VLAN settings in your networking environment.
TR069	To display and configure TR069 settings. ● Active: To enable or disable TR069 feature. ● Version: To select supported TR069 version (version 1.0 or 1.1). ● ACS/CPE: ACS is short for Auto configuration servers as server side, CPE is short for Customer-premise equipment as client side devices. ● URL: To configure URL address for ACS or CPE. ● User name: To configure username for ACS or CPE. ● Password: To configure Password for ACS or CPE. ● Periodic Inform: To enable periodically inform. ● Periodic Interval: To configure interval for periodic inform. **Note**: TR-069(Technical Report 069) is a technical specification entitled CPE WAN Management Protocol (CWMP).It defines an application layer protocol for remote management of end-user devices.

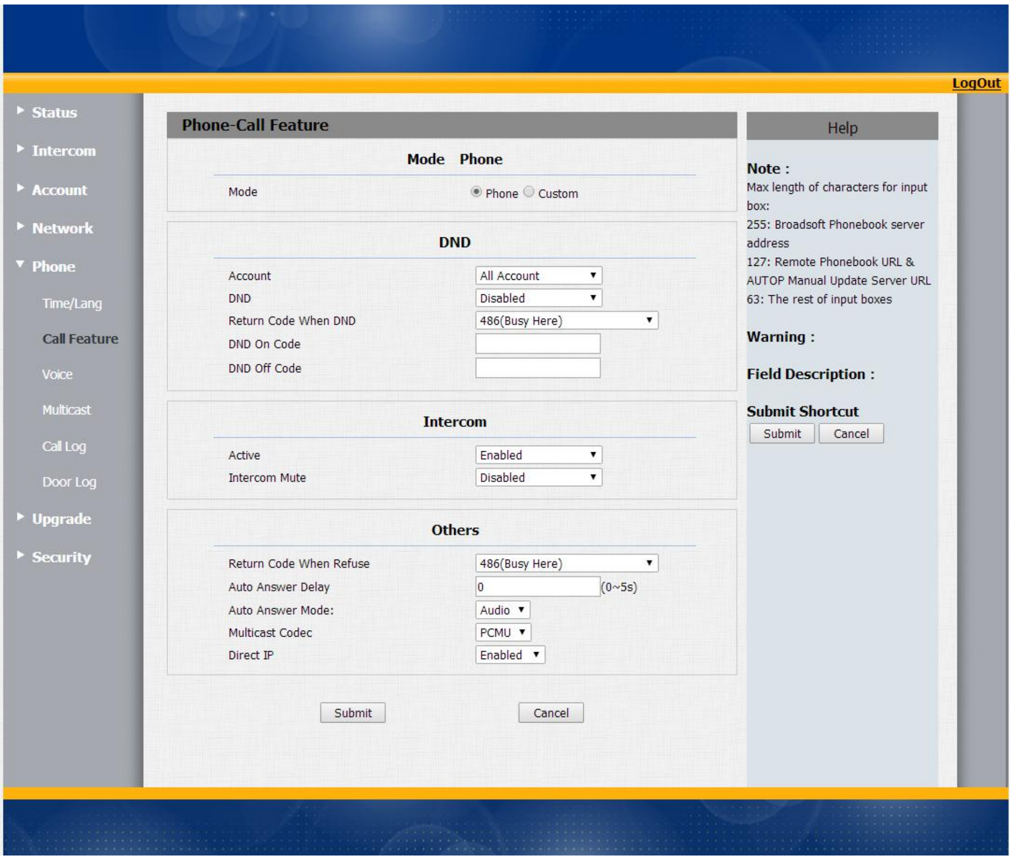
2.6. Phone

2.6.1 Time/Lang



Sections	Description
NTP	To configure NTP server related settings. ● Time Zone: To select local Time Zone for NTP server. ● Primary Server: To configure primary NTP server address. ● Secondary Server: To configure secondary NTP server address, it takes effect if primary NTP server is unreachable. ● Update interval: To configure interval between two consecutive NTP requests. **Note**: NTP, Network Time Protocol is used to automatically synchronized local time with INTERNET time, since NTP server only response GMT time, you need to specify the Time Zone for IP phone to decide the local time.

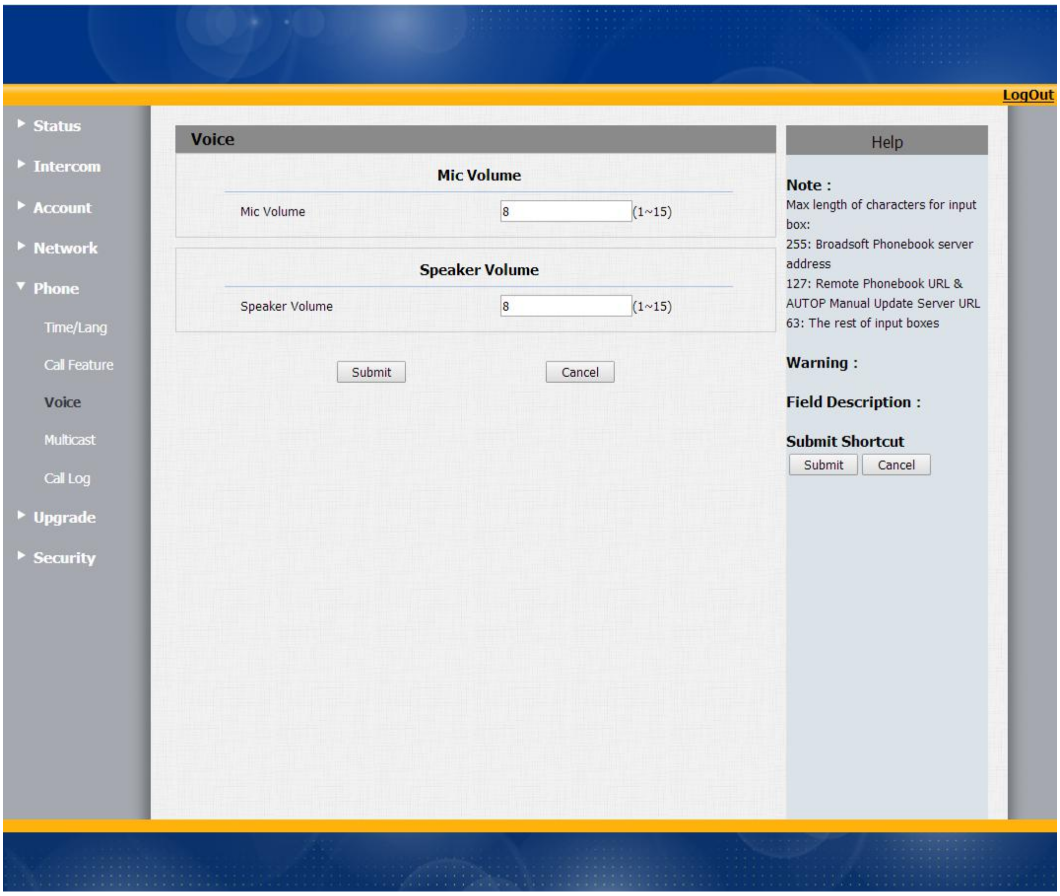
2.6.2 Call Feature



Sections	Description
Mode	● Mode: Select the desired mode.
DND	DND (Do Not Disturb) allows IP phones to ignore any incoming calls. ● Return Code when DND: Determine what response code should be sent back to server when there is an incoming call if DND on. ● DND On Code: The Code used to turn on DND on server's side, if configured, IP phone will send a SIP message to server to turn on DND on server side if you press DND when DND is off. ● DND Off Code: The Code used to turn off DND on server's side, if configured, IP phone will send a SIP message to server to turn off DND on server side if you press DND when DND is on.
Intercom	Intercom allows user to establish a call directly with the callee. ● Active: To enable or disable Intercom feature. ● Intercom Mute: If enabled, once the call established, the

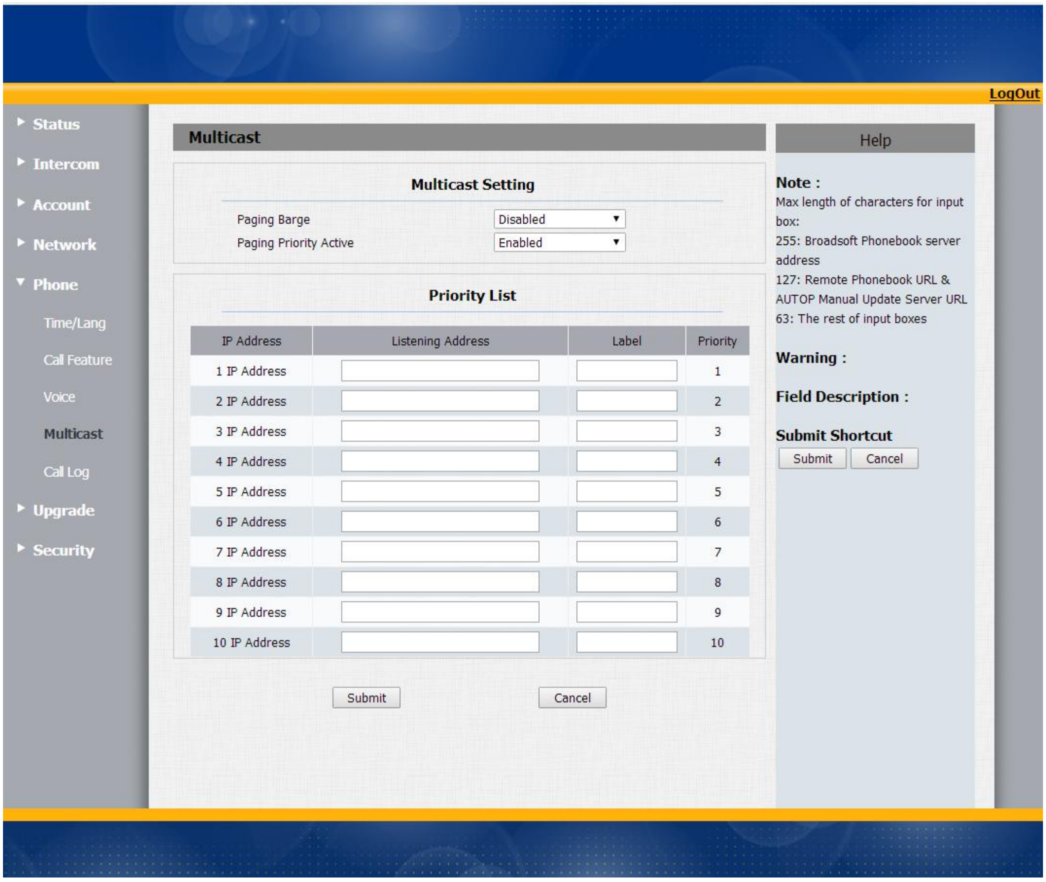
	callee will be muted.
Others	● Return Code When Refuse: Allows user to assign specific code as return code to SIP server when an incoming call is rejected. ● Auto Answer Delay: To configure delay time before an incoming call is automatically answered. ● Auto Answer Mode: To set video or audio mode for auto answer by default. ● Direct IP: Direct IP call without SIP proxy.

2.6.3 Voice



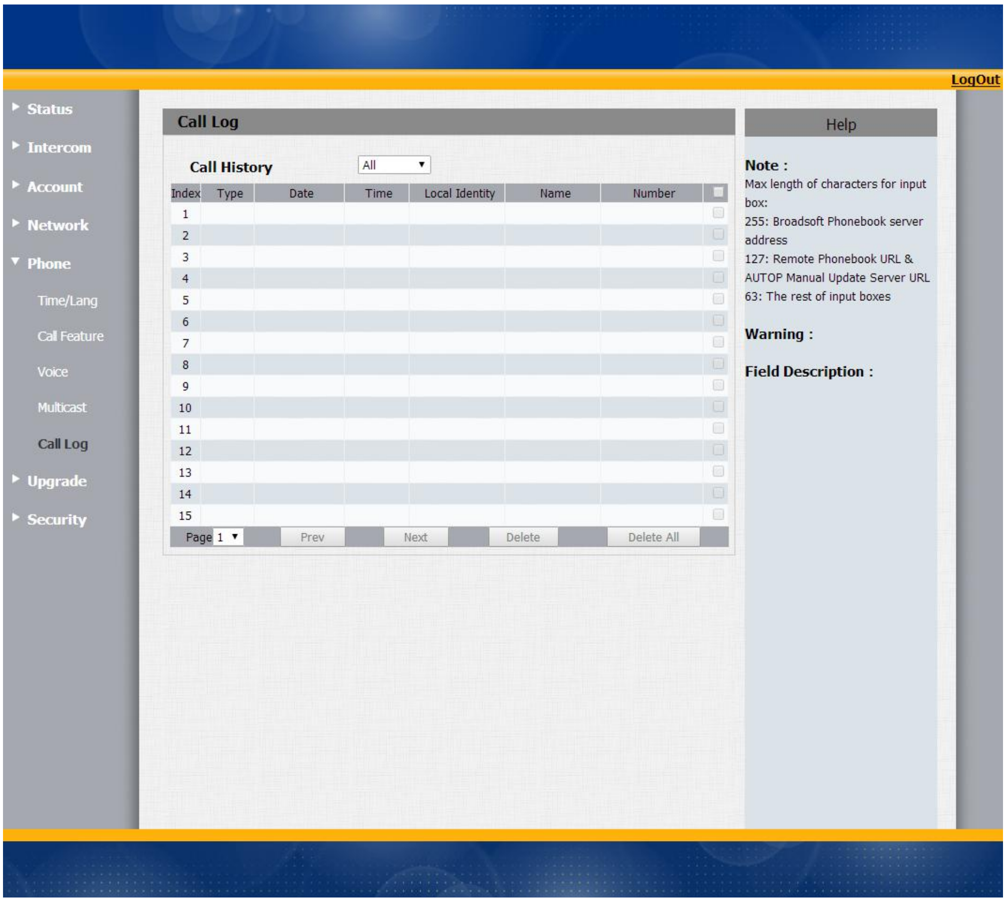
Sections	Description
Mic Volume	To configure Microphone volume , from 1-15,8 by default.
Speaker Volume	To configure Speaker Volume,from 1-15,8 by default.

2.6.4 Phone-Multicast



Sections	Description
Multicast Setting	To display and configure the Multicast setting. ● Paging Barge: Choose the multicast number ,range from 1 to 10. ● Paging priority Active: Enable o disable the multicast.
Priority List	To setup the multicast parameters. ● Listening Address: Enter the IP address you need to listen ● Label： Input the label for each listening address

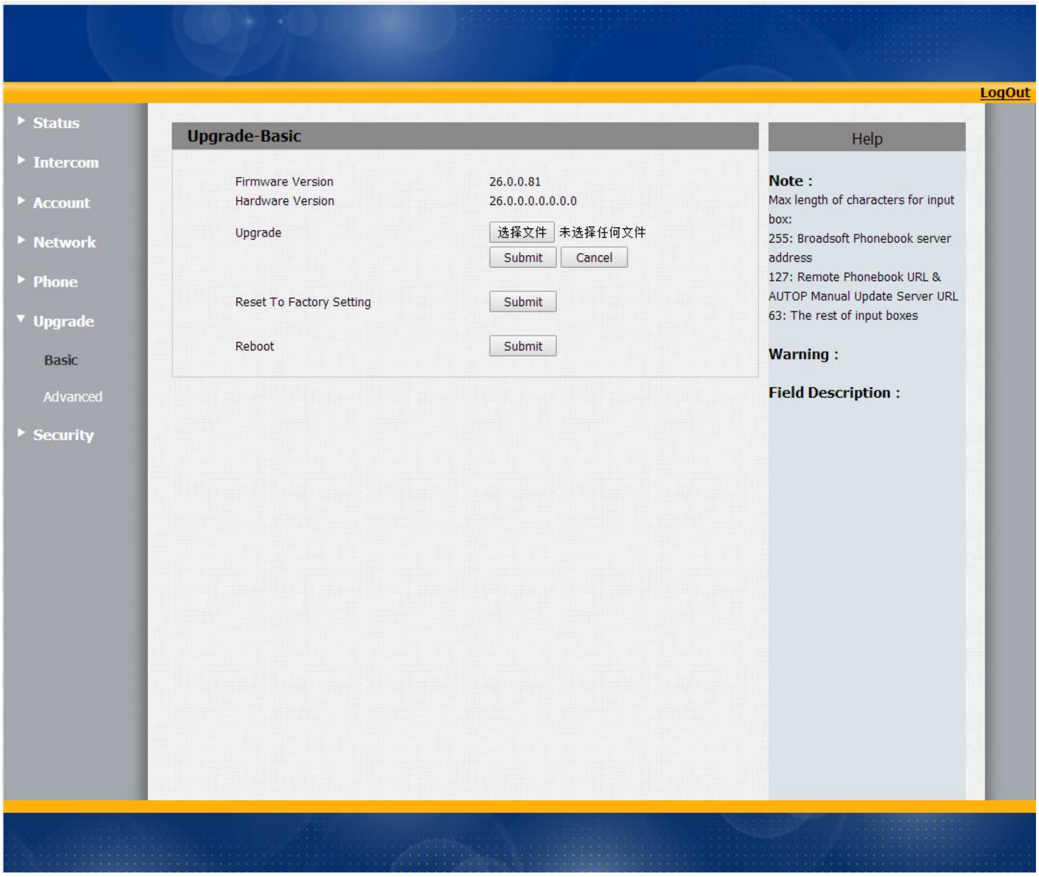
2.6.5 Phone-Call Log



Sections	Description
Call History	To display call history records. Available call history types are All calls, Dialed calls, Received calls, Missed calls, Forwarded calls. Users can check the call history in detail. Tick the number to delete or delete all logs. R23X supports 100 call logs.

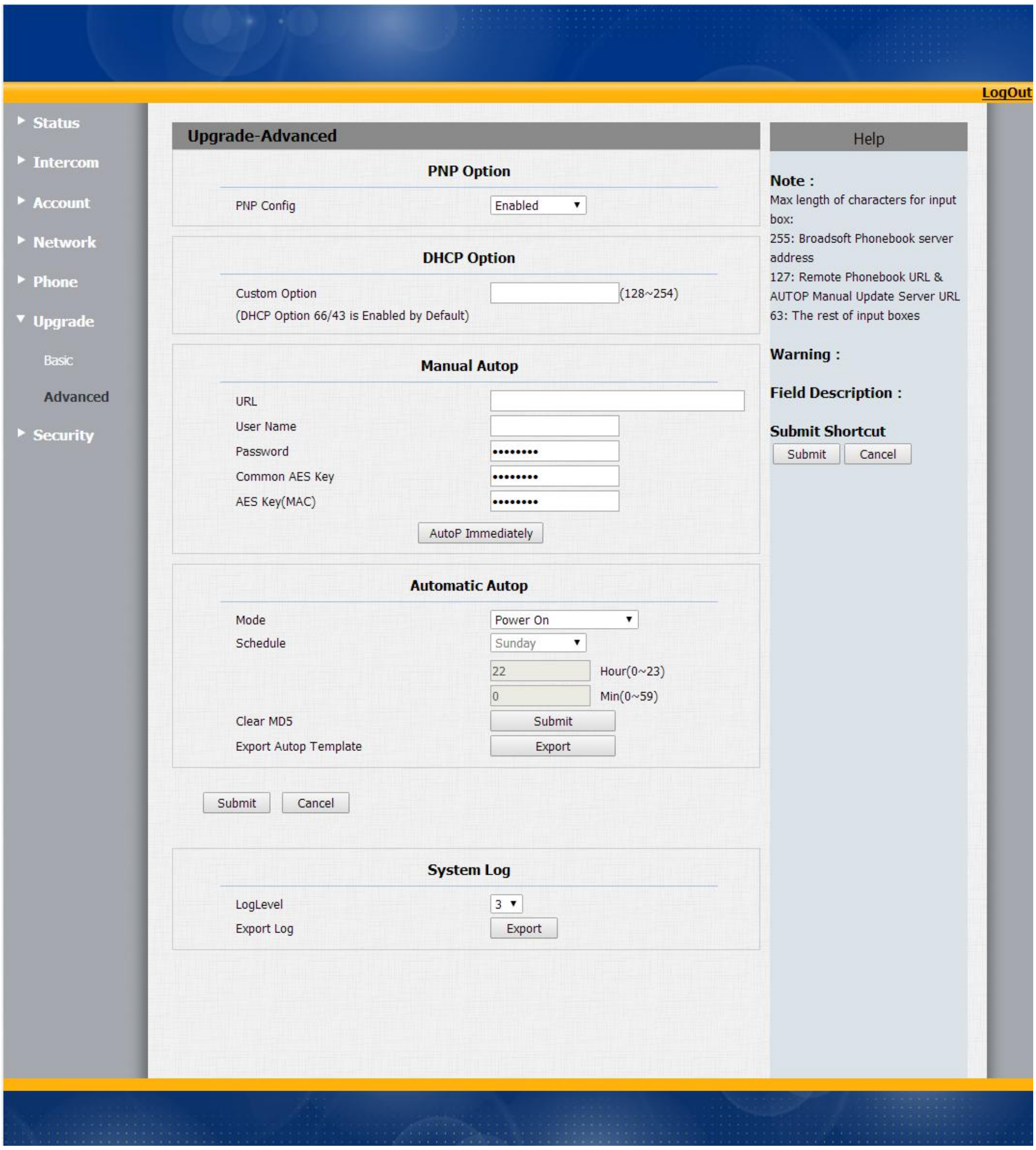
2.7. Upgrade

2.7.1 Basic



Sections	Description
Upgrade	To select upgrading zip file from local or a remote server automatically. **Note:** Please make sure it's right file format for right model.
Firmware version	To display firmware version, firmware version starts with MODEL name.
Hardware Version	To display Hardware version.
Reset to Factory Setting	To reset IP phone's setting to factory settings.
Reboot	To reboot IP phone remotely from Web UI.

2.7.2 Advance

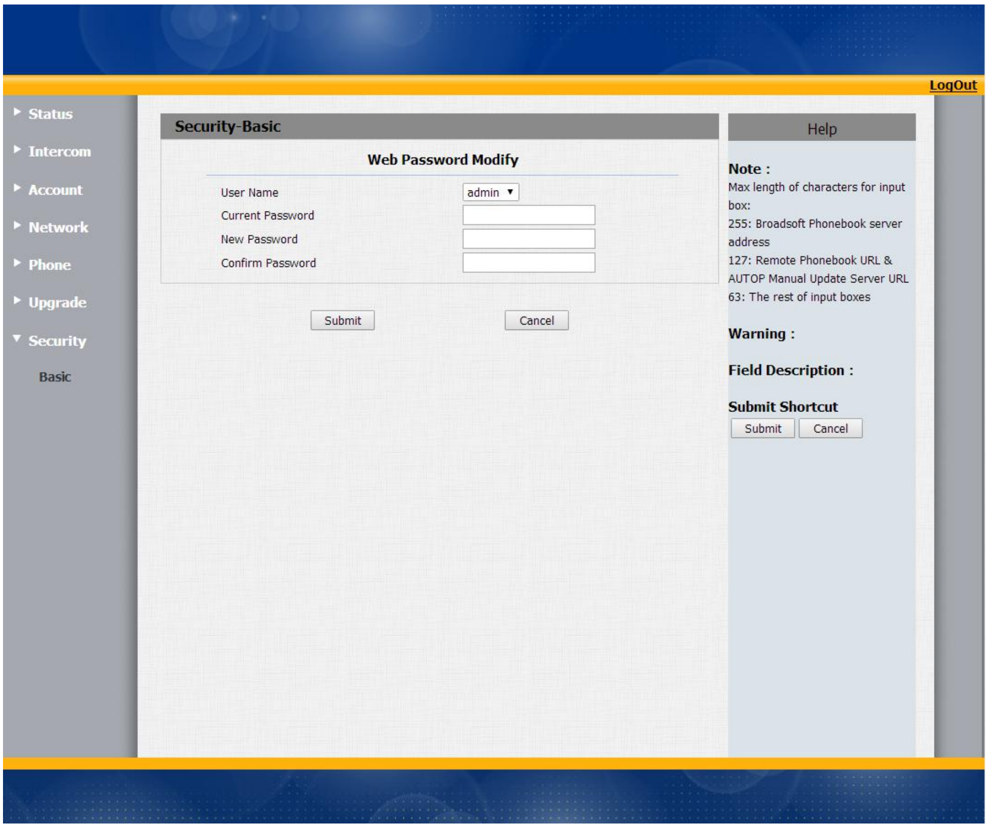


Sections	Description
PNP Option	To display and configure PNP setting for Auto Provisioning. ● PNP: Plug and Play, once PNP is enabled, the phone will send SIP subscription message to PNP server automatically to get Auto Provisioning server's address. By default, this SIP message is sent to multicast address 224.0.1.75(PNP server address by standard).
DHCP Option	To display and configure custom DHCP option. ● DHCP option: If configured, IP Phone will use designated DHCP option to get Auto Provisioning server's address via DHCP.

	This setting require DHCP server to support corresponding option.
Manual Update Server	To display and configure manual update server's settings. ● URL: Auto provisioning server address. ● User name: Configure if server needs an username to access, otherwise left blank. ● Password: Configure if server needs a password to access, otherwise left blank. ● Common AES Key: Used for IP phone to decipher common Auto Provisioning configuration file. ● AES Key (MAC): Used for IP phone to decipher MAC-oriented auto provisioning configuration file(for example, file name could be 0c1105888888.cfg if IP phone's MAC address is 0c1105888888). **Note:** AES is one of many encryption, it should be configure only configure file is ciphered with AES, otherwise left blank.
AutoP	To display and configure Auto Provisioning mode settings. This Auto Provisioning mode is actually self-explanatory. For example, mode "Power on" means IP phone will go to do Provisioning every time it powers on.
System Log	To display system log level and export system log file. ● System log level: From level 0~7.The higher level means the more specific system log is saved to a temporary file. By default, it's level 3. ● Export Log: Click to export temporary system log file to local PC.

2.8. Security

2.8.1 Basic



Sections	Description
Web Password Modify	To modify user's password. ● Current Password: The current password you used. ● New Password: Input new password you intend to use. ● Confirm Password: Repeat the new password.